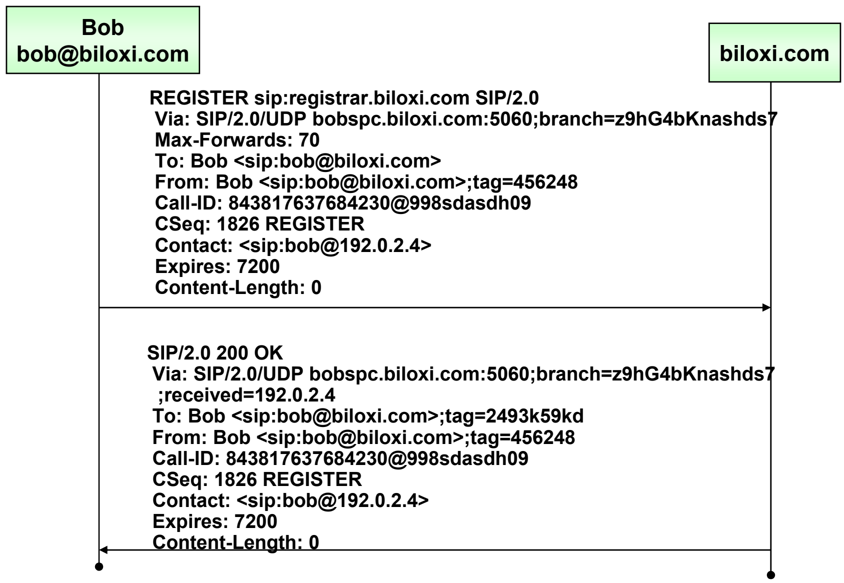
Call Setup Examples based on Generic SIP

Registration example with SIP



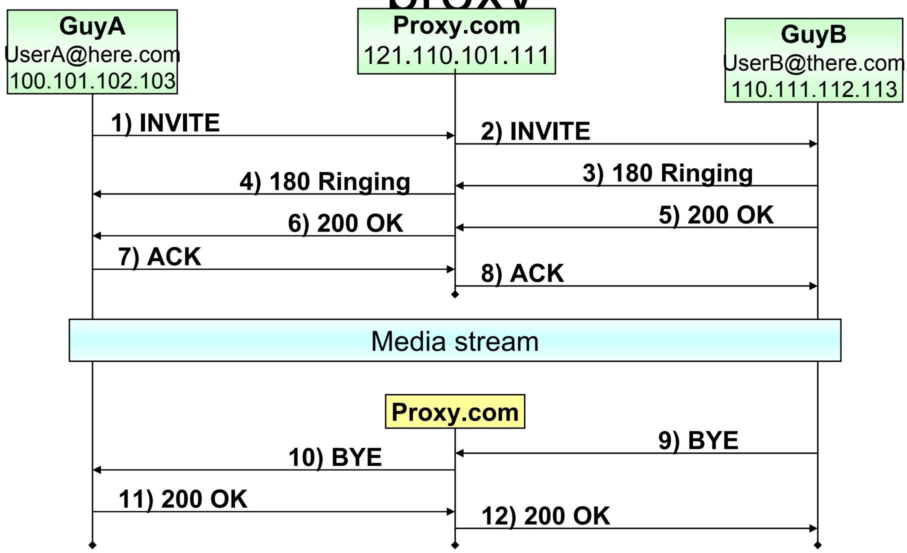
In this example Bob’s User Agent performs a successful registration to a registrar whose domain name is biloxi.com.

Registration is set to expire after two hours (7200 seconds). It may be seen in header Expires:

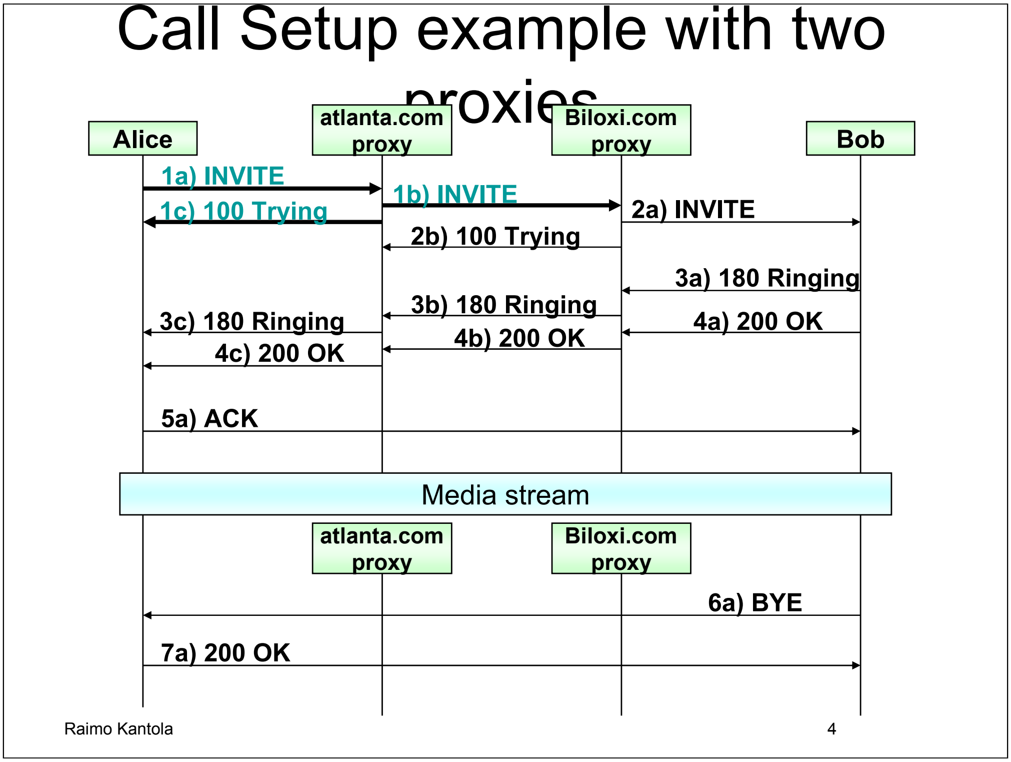
Registrar answers with 200 OK response, meaning that registration was succesfull.

(example from RFC3261)

Call Setup example with one proxy



In this example GuyA (UserA@here.com at 100.101.102.103) performs a successful call (session) initiation to GuyB (UserB@there.com at 100.101.102.103). For the sake of simplicity in the example both party is registered the same SIP proxy (proxy.com).



In this example Alice, who is registered to atlanta.com, performs a successful call (session) initiation to Bob, who is registered to biloxy.com.

<u>1a) Alice -> atlanta.com proxy</u>

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

(Alice's SDP not shown)

Alice's UA sets Max-Forwards to 70. INVITE is addressed to logical SIP address, and Alice relies on proxy to find Bob. Alice is currently at pc33.atlanta.com and she puts current address in header Contact:

<u>1b) INVITE atlanta.com proxy -> biloxi.com proxy</u>

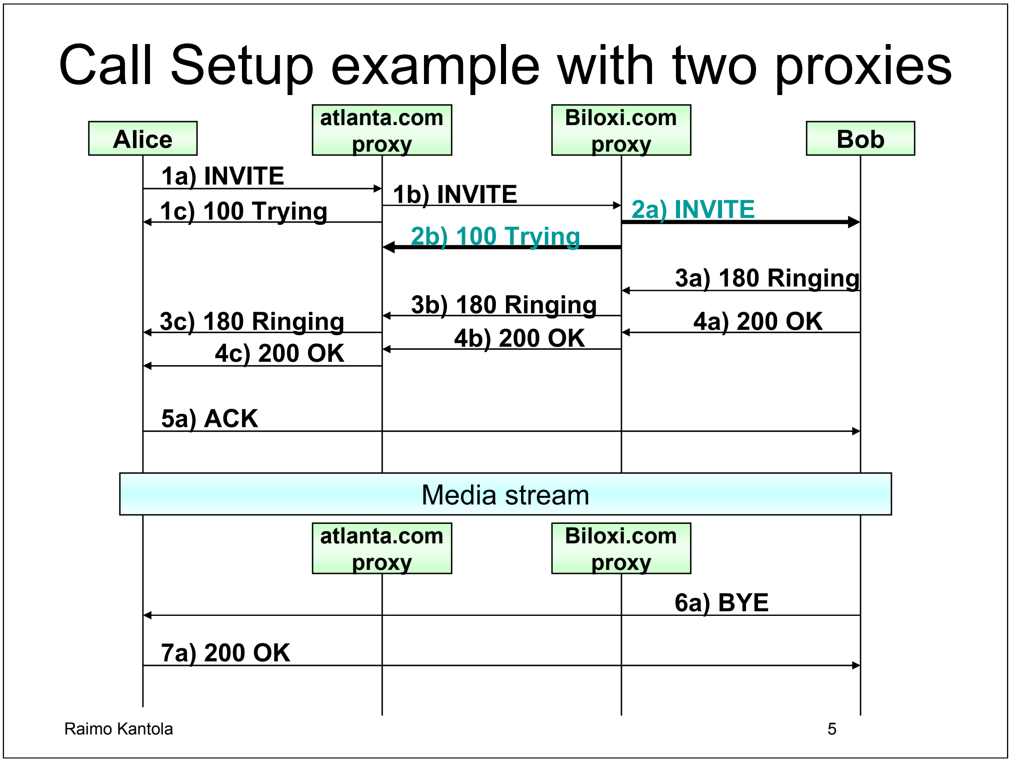
```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
Max-Forwards: 69
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

(Alice's SDP not shown)

Atlanta.com proxy decreases Max-forwards by 1. Adds Via: header (puts its own address and branch there)

Atlanta.com leaves all other headers unchanged and resolves Bob's domain proxy. It then sends INVITE to biloxi.com proxy

After atlanta.com has forwarded INVITE to biloxi.com proxy, it sends 100 Trying to Alice. It will make her User



In this example Alice, who is registered to atlanta.com, performs a successful call (session) initiation to Bob, who is registered to biloxy.com.

2a) biloxi.com proxy -> Bob

```
INVITE sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP server10.biloxi.com;branch=z9hG4bK4b43c2ff
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
Max-Forwards: 68
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142
```

Biloxi.com proxy decreases max-forwards by 1. It also adds its address in Via: header, and assigns branch to it.

Note that in first line there is Bob's current address (sip:bob@192.0.2.4). It has been obtained from Location server.

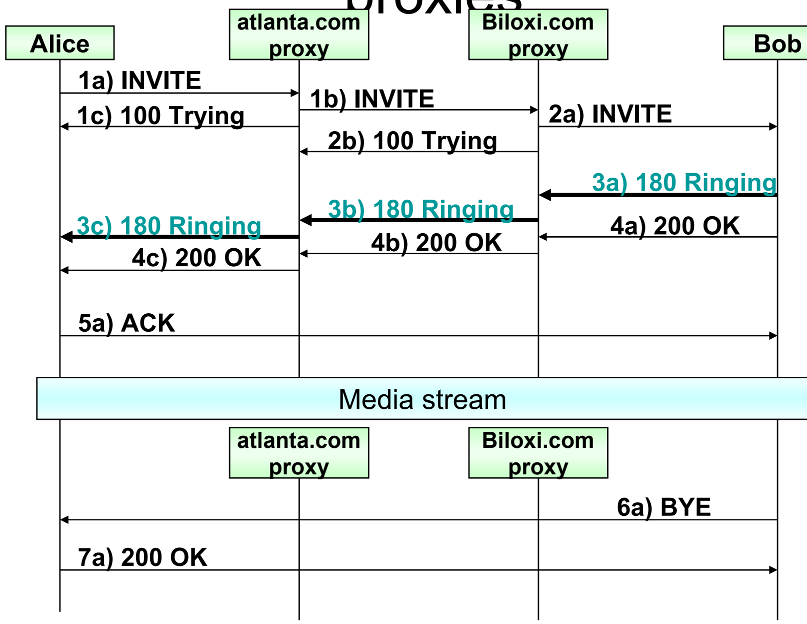
(Alice's SDP not shown)

2b) TRYING biloxi.com proxy -> atlanta.com proxy

After biloxi.com has forwarded INVITE to Bob, it sends 100 Trying to átlanta.com proxy.

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
Max-Forwards: 69
To: Bob <sip:bob@biloxi.com>
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:alice@pc33.atlanta.com>
```

Call Setup example with two proxies



In this example Alice, who is registered to atlanta.com, performs a successful call (session) initiation to Bob, who is registered to biloxy.com.

3a) Bob -> biloxi.com proxy

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.0.2.4>
CSeq: 314159 INVITE
Content-Length: 0

Bob's UA sends 180 Ringing provisional response. It will send it back via the same route (usin Via: headers). Response will go to the topmost Via: address.

Also, Bob's UA adds tag to To: header. Now dialog is completely determined (with Call-ID, From: and To: tags)

3b) biloxi.com proxy -> atlanta.com proxy

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com
;branch=z9hG4bKnashds8 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
Contact: <sip:bob@192.0.2.4>
CSeq: 314159 INVITE
```
Content-Length: 0

From biloxi.com proxy to atlanta.com proxy.

3c) atlanta.com proxy -> Alice

Alice receives 180 Ringing response. Now Alice's UA knows that Bob has been alerted.

Call Setup example with two proxies

4a) Bob -> biloxi.com proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP server10.biloxi.com
;branch=z9hG4bK4b43c2ff8.1;received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com
;branch=z9hG4bK77ef4c2312983.1 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

(Bob's SDP not shown)

Bob accepts invitation and sends 200 OK final response. It will go through the same set of proxies (determined by Via: headers)

Bob puts his current addres in Contact: header. Alice is now able to contact him directly. Subsequent SIP messages may go directly to Bob, and not through proxies

4b) biloxi.com proxy -> atlanta.com proxy

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com;branch=z9hG4bK77
 ;received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds8
 ;received=192.0.2.1
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

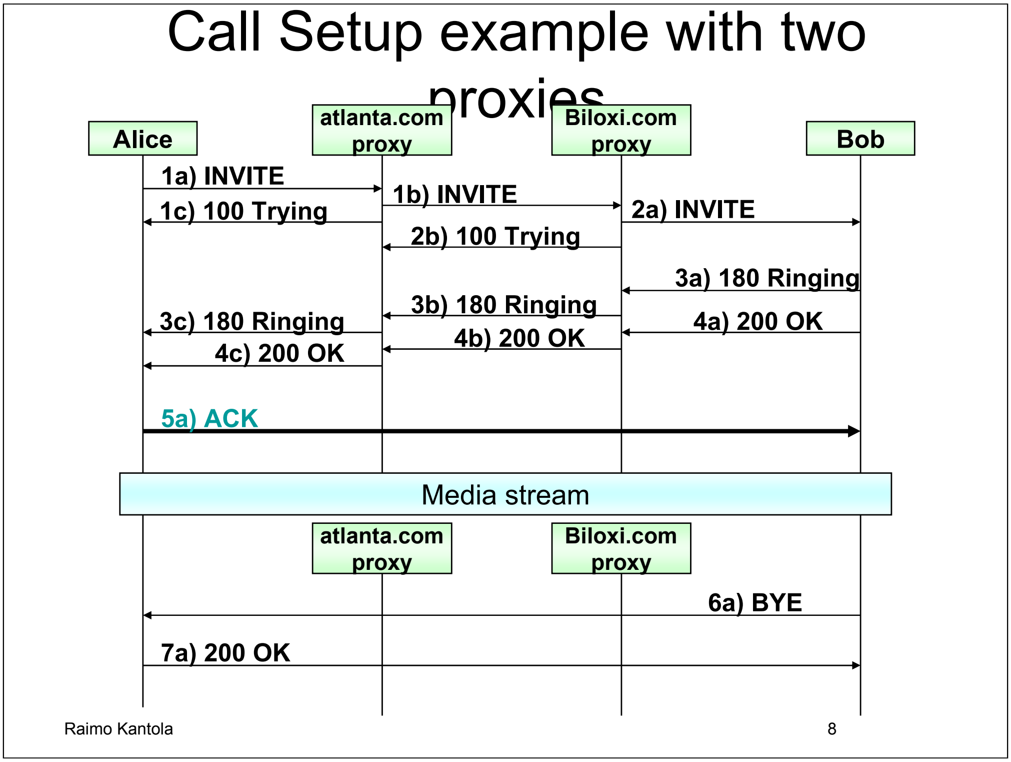
(Bob's SDP not shown)

Biloxi.com proxy forwards response to topmost via: header address. (it removed its own address in Via: header previously)

4c) 100 Trying atlanta.com proxy -> Alice

```
SIP/2.0 200 OK
```

Alice receives 200 OK with Bob's session parameters in the message body (not shown here). If Alice can accept it, she will send ACK message



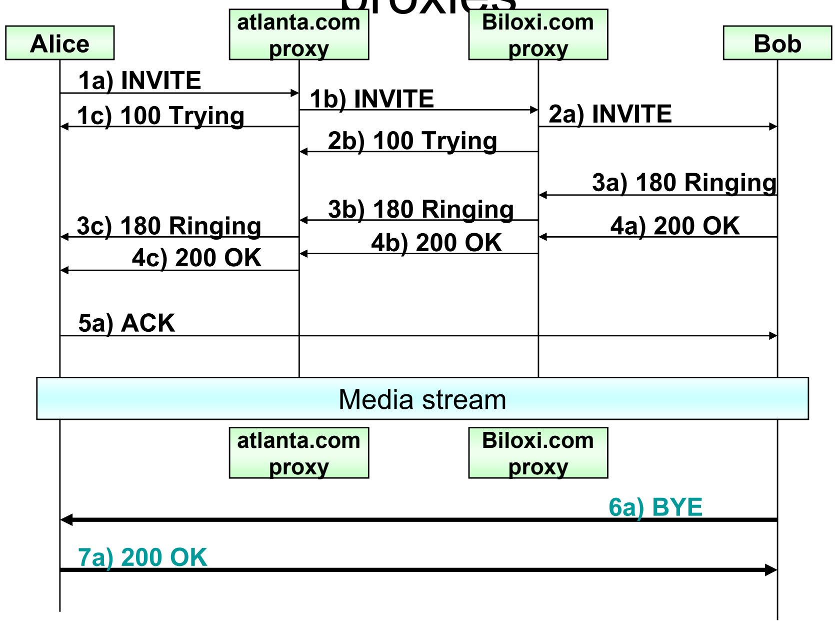
In this example Alice, who is registered to atlanta.com, performs a successful call (session) initiation to Bob, who is registered to biloxy.com.

5a) Alice -> Bob

```
ACK sip:bob@192.0.2.4 SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bKnashds
Max-Forwards: 70
To: Bob <sip:bob@biloxi.com>;tag=a6c85cf
From: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 ACK
Content-Length: 0
```

The media session between Alice and Bob is now established. They agreed on session parameters (described in SDP body)

Call Setup example with two proxies



6a) Bob -> Alice

```
BYE sip:alice@pc33.atlanta.com SIP/2.0
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnashds10
Max-Forwards: 70
From: Bob <sip:bob@biloxi.com>;tag=a6c85cf
To: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

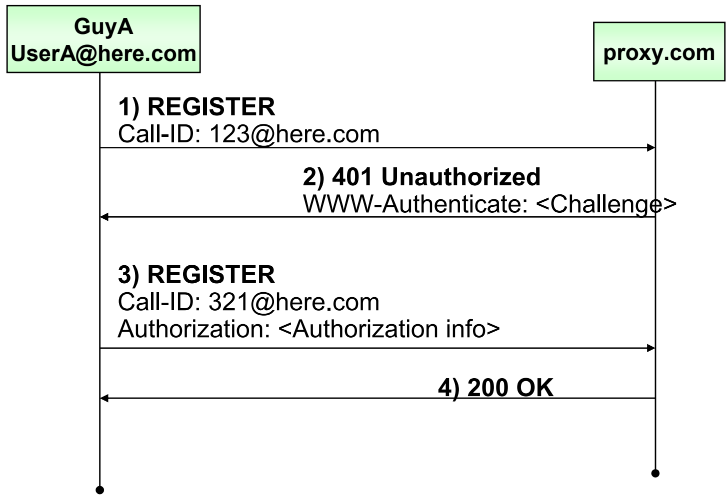
Bob after a while decides to disconnect. Bob’s UA has its own CSeq sequencing (note) and From: and To: fields are swapped, because Bob is originating a request. But Bob still refers to the same dialog (can be seen by Call-ID and tags)

7a) Bob -> Alice

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.0.2.4;branch=z9hG4bKnashds10
From: Bob <sip:bob@biloxi.com>;tag=a6c85cf
To: Alice <sip:alice@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 231 BYE
Content-Length: 0
```

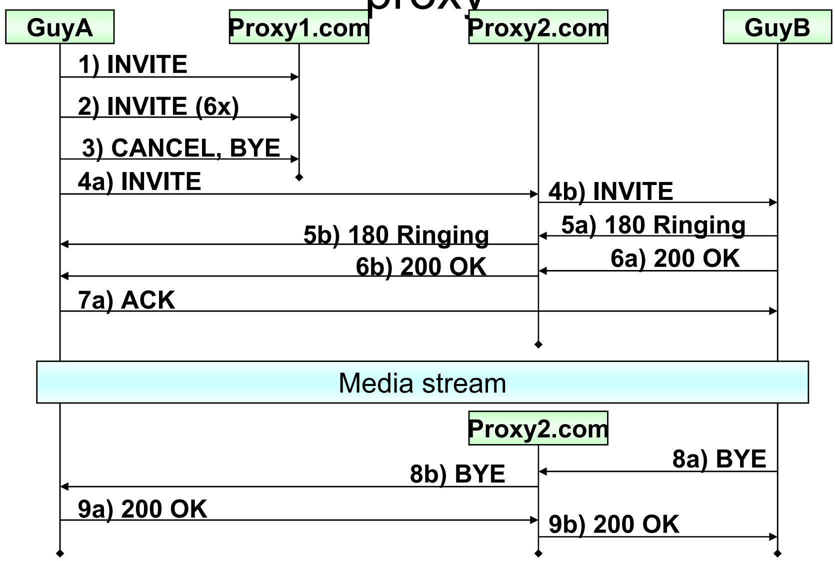
Alice acknowledges BYE, and call is over. This 200 OK will refer to the BYE request, and that can be seen from CSeq field, carrying BYE method name

Registration example with SIP authentication

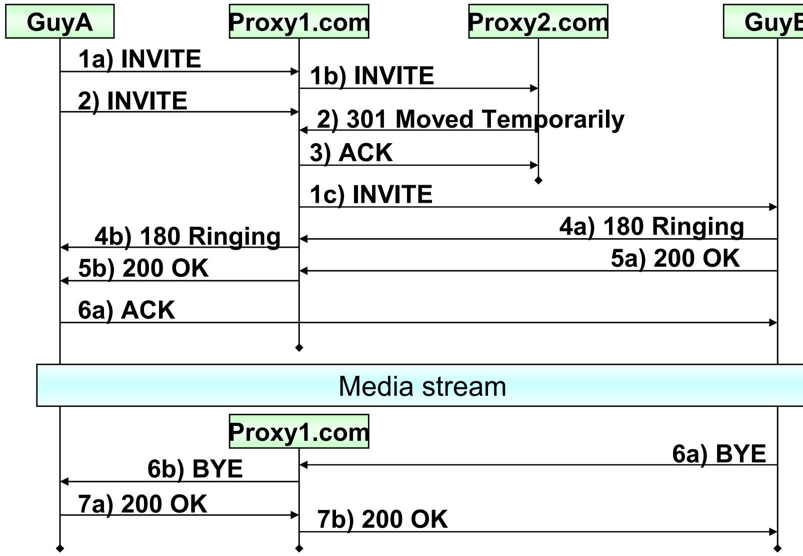


In this example GuyA (UserA@here.com at 100.101.102.103) performs a successful registration to a proxy whose domain name is proxy.com.

Call Setup example with a non-working proxy



Call Setup example with a Redirect server

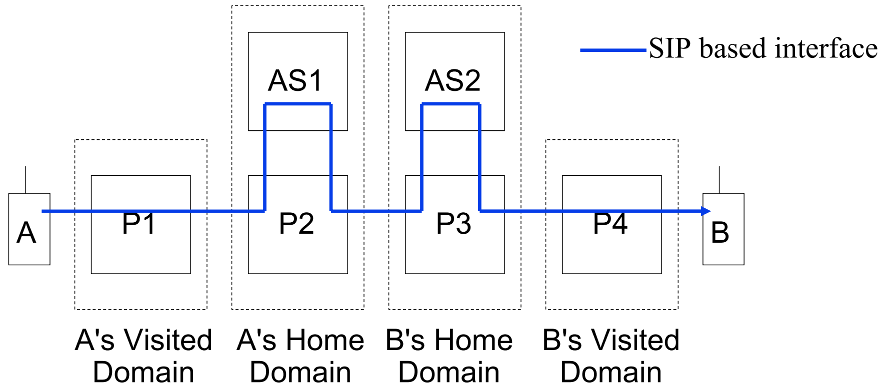


In this example GuyA, who is registered to proxy1.com, performs a successful call (session) initiation to GuyB, who is registered to proxy2.com.

Services use many protocols

- New services and more flexible service creation should differentiate IP Communications Network from PSTN
- Services should combine different forms of communication, thus multiple protocols are needed:
  - SIP for media sessions and session related services, subscriptions and notifications?, messaging?
  - HTTP for web and transactions
  - SMTP for e-mail
  - RTSP for media streaming
- The use of these protocols is orchestrated by the service logic: context is set up using SIP.

Routing and Service Model in 3G



P1, P4: Outbound Proxies

P2, P3: Registrar Proxies

AS1, AS2: Application Servers

NB: Also AS based on direct processing of call state: There is no Basic call state model like in IN

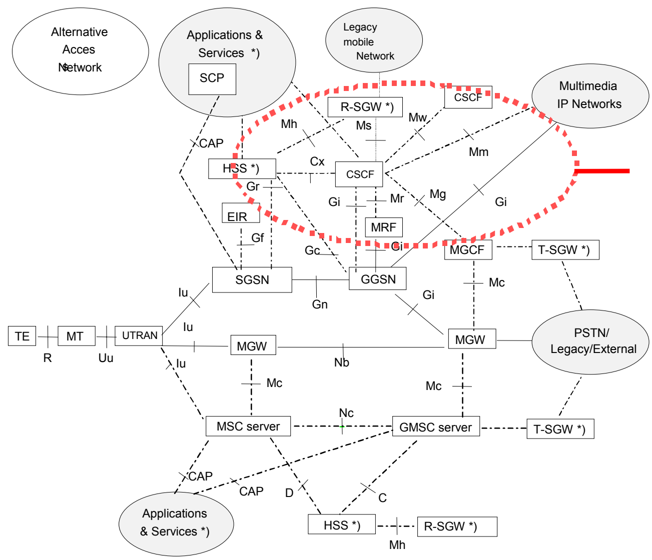
SIP Entities & Service Capabilities in IMS

- User Agent (= UAC + UAS)
  - Can run services, such as forwarding, filtering etc.
  - Not always connected (out of coverage/battery etc.)
- Redirect Server
  - Can do services that require only Request-URI change, e.g. translation, parameter addition etc.
- Proxy Server
  - Can change certain headers and stay in the signaling path
  - Forking, actions based on responses
- Back-to-Back User Agent (=both ways User Agent)
  - Can e.g. issue requests to a call leg or modify SDP, generate ACK and 200OK, like UAC/UAS
  - In many cases necessary e.g Sessio Border Controller or 3G SIP

Application Server in 3G

- Fuzzy Definition but has SIP+ interface!
- Can be a Redirect or Proxy Server or Back-to-Back UA
- The key is that it should be *programmable*
  - Routing based on service logic: what to do when user not registered or busy
  - URI translation: Reachability chains
  - Interfaces to other protocols: HTTP, SMTP, RTSP etc.
- Can be single purpose boxes, or multi-purpose boxes, or controllers that orchestrate things

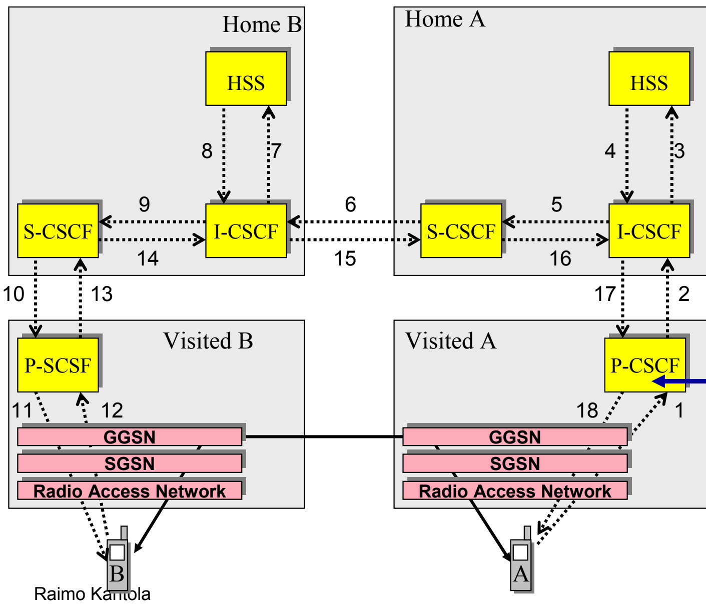
3GPP Network Model (preliminary: …



CSCF (Call/Session Control Function) is the primary SIP node in the network.

(from www.sipforum.org)

Different Kinds of CSCFs



Proxy CSCF:
In the same network as the GGSN: if the visited net does not support IMS, can be in the home network.

Provides
-emergency service breakout,
- triggers for locally-provided services, and
- number normalizing (per local dialing plan)
- Policy Decision point

-user authentication

-maintains a security association with the terminals for signaling

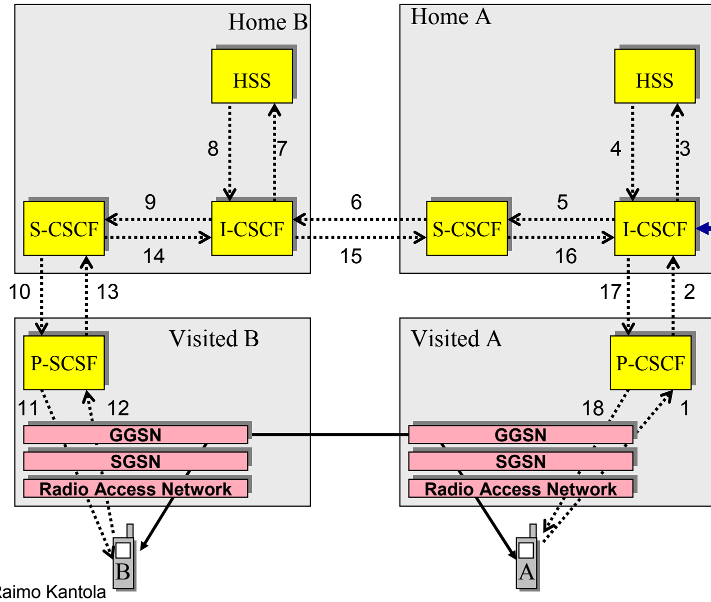
Currently, 3GPP has defined three different functional behaviors which the CSCF will exhibit.

The Proxy CSCF (P-CSCF) provides a first point of contact for the handset. All signaling to and from the handset goes through the P-CSCF. In terms of SIP, it behaves as an outbound proxy.

The main purpose for this node is to provide emergency service breakout and to do some basic message manipulation to enable the visited domain operator to provide locally sensitive services (e.g. traffic reports, directory services, etc). It also does simple number internationalization (which allows the support of local dialing plans).

It will probably also play a role in quality of service reservations.

Different Kinds of CSCFs



Interrogating CSCF: Queries the HSS to find the correct S-CSCF. First point of contact for incoming call signalling.

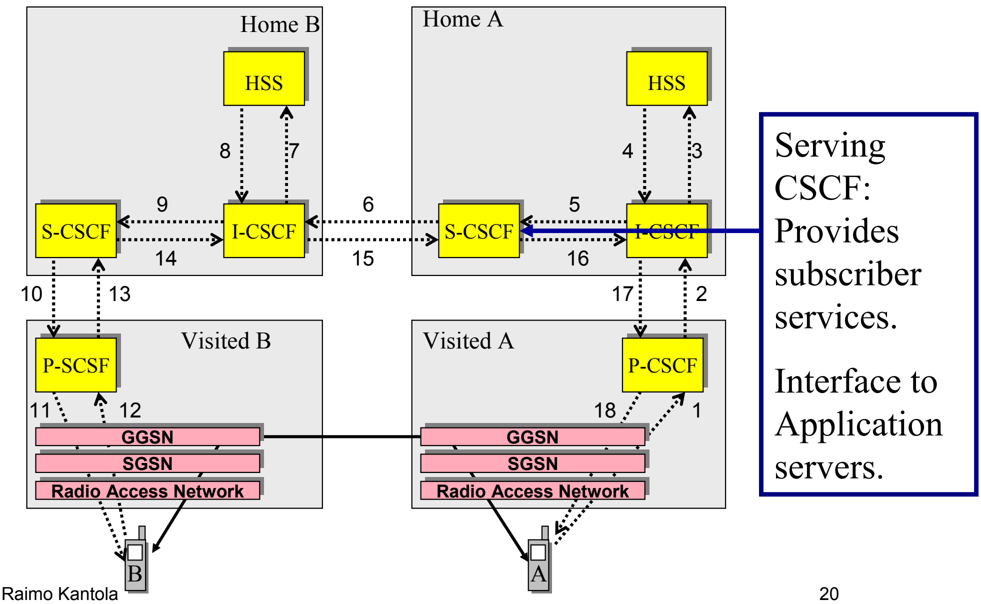
Load distribution node!

The Interrogating CSCF (I-CSCF) is mostly a load distribution node. Since DNS allows us simple statistical distribution among identical nodes, distributing load among the I-CSCFs is quite simple. But if all we relied on was statistical distribution, we wouldn't be able to allocate subscriptions on appropriate serving nodes according to their capabilities, nor would we be assured of the ability to keep call state information between transactions.

So, the I-CSCF, in conjunction with the HSS, allocates subscription information onto appropriate Serving CSCFs. The HSS keeps track of this information so that all transactions and all calls for the same user go through the same service node.

The HSS stores user profile information; it's somewhat similar to the HLR found in today's cellular networks.

Different Kinds of CSCFs



Raimo Kantola

The Serving CSCF (S-CSCF), quite simply, provides users services.

Of course, SIP allows the terminal to provide many services itself. The S-CSCF will be useful in providing, for example: call forwarding when the terminal is not available, call barring, centralized speed dial lists, VPN services, etc.

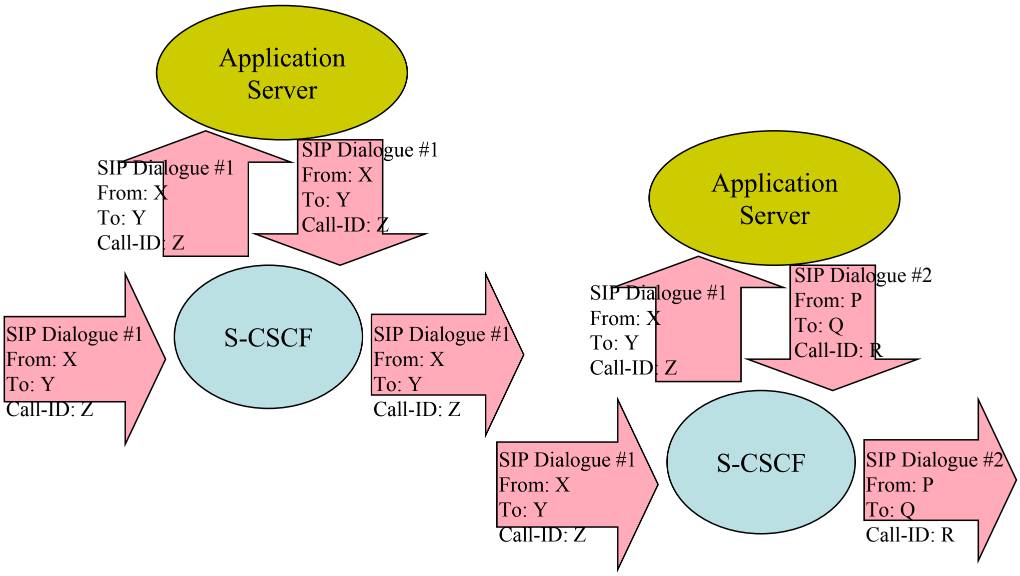
The interface to AS is based on iFC (initial Filter Criteria) and SIP. The interface also has a historic label ISC (IMS Service Control) and ”possibly SIP with some extensions”. In practice the protocol for AS communication is pure SIP. When a filter matches, the system finds the address of the AS that needs to get involved in proving the service.

The AS can a UA, a SIP proxy, a SIP redirect Server or a B2BUA = a collection of UAs with some service logic binding them together.

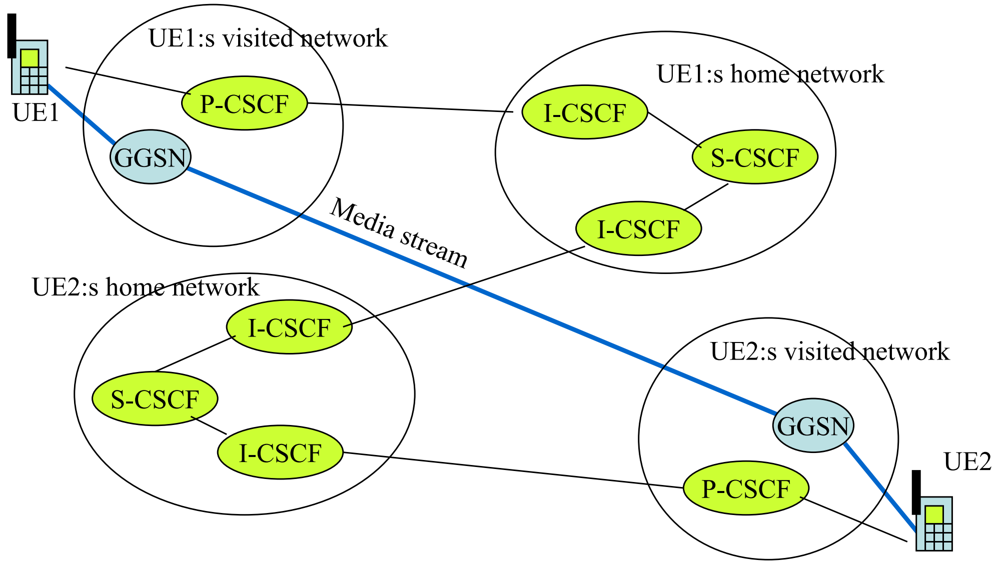
To route the call to the AS the S-CSCF creates a ROUTE header with two entries (or adds two new entries into the ROUTE header) containing the SIP URI of the AS and its own SIP URI in the second place. Based on the latter the AS will know that it needs to route the request back to the S-CSCF. The own SIP URI also contains some state info in the username part of the URI. When the request returns to the S-CSCF, it uses this state info to figure out where to continue the call processing.

The AS may or may not decide to stay on the signaling path. To stay on the path, the AS places its SIP URI in the RECORD ROUTE header

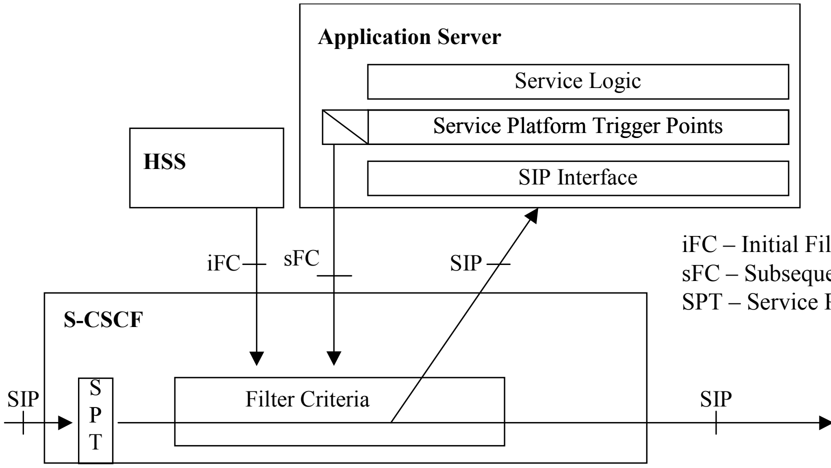
SIP Proxy vs B2BUA



Overview of routing between two mobile terminals



3G Application Triggering



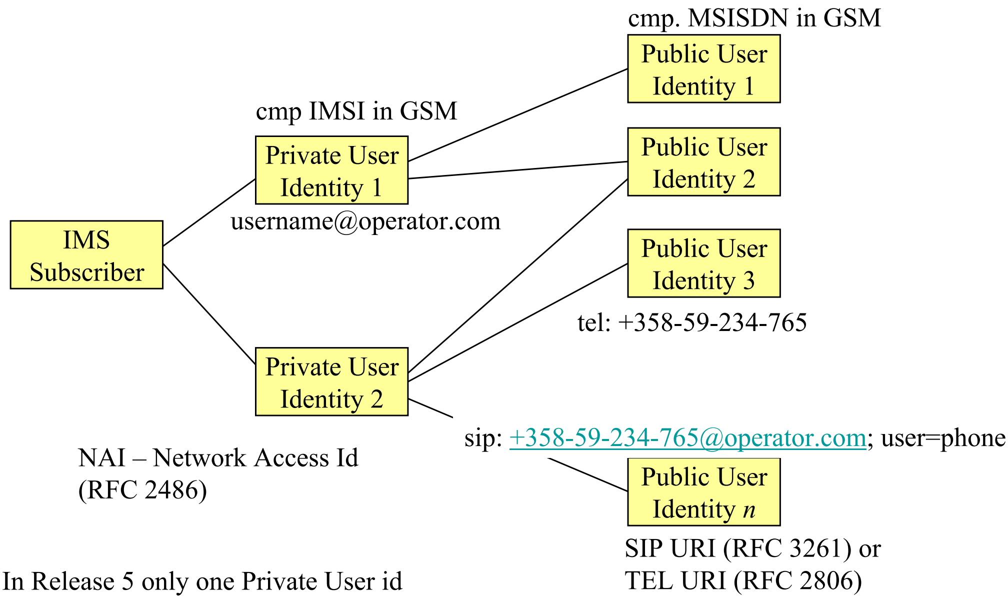
iFC – Initial Filter Criteria
sFC – Subsequent Filter Criteria
SPT – Service Point Trigger

Service processing can be delegated to Application Servers with a fine grained control:
Filter criteria in IMS triggering is bound to user identities, since a user may have many identities, different services may be invoked depending on the identity.

The originating IMS terminal sets the Preferred Identity for application triggering in the P-Preferred-Identity header field. The P-CSCF verifies that this a legal identity for the particular user within the current security association, changes the header field to P-Asserted-Identity with the value from the P-Preffered-Identity field. If the verification fails, the P-CSCF chooses to forward the default user identity in the P-Asserted-Identity field. If there was no P-Preferred-Identity in the INVITE, P-CSCF will insert the default user id into the P-Asserted-Identity header field.

Another factor that may serve as criteria for certain services is the type of access network (ADSL, WLAN, GERAN, UTRAN etc) in the P-Access-Network-Info. This information is carried only until the calling user’s home network and never forwarded into the callee’s home network for privacy reasons. The type of the access network gives an idea about the available capacity and pricing for the capacity.

Identification of users in 3G IMS in R6



In Release 5 only one Private User id

How to Program Services

- Call Processing Language
- SIP CGI
- SIP Servlets
- SIP JAIN (JSLEE – Jain Serv Logic Exec Env)
- Soft SSF and INAP/CAP
- Parlay
- OSA

=> Whatever… Different abstraction levels

==>
There will be many competing ways to implement services!

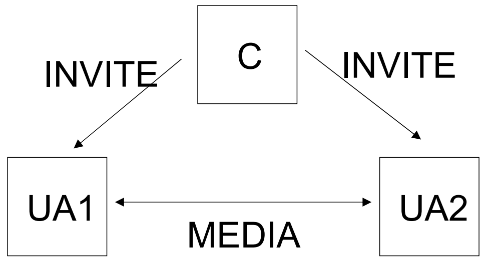
The claim is that it should be as open as flexible as creating services in the web these days

Server types for different services

- Media Server (SIP, RTSP, HTTP)
  - Announcements, IVR, Voicemail, Media on demand
- Conferencing Server (SIP)
  - Media mixer
- Presence Server (SIP)
  - Users status info, capabilities, willingness to communicate
- Web Server (HTTP), E-mail Server (SMTP), Messaging Server (SIP?), Text-to-Speech Server etc.
- Controller Server
  - Co-ordinates the overall service

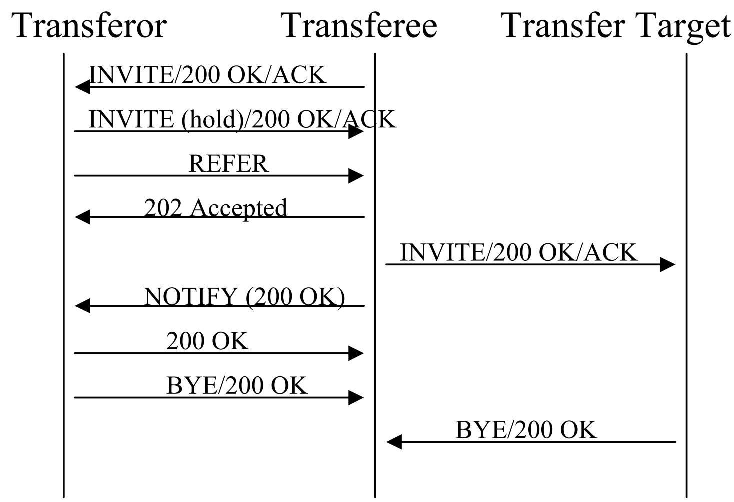
=> Server resources can be addressed by URLs, no need for tight coupling a la MGCP/Megaco

Third Party Call Control is based on SIP



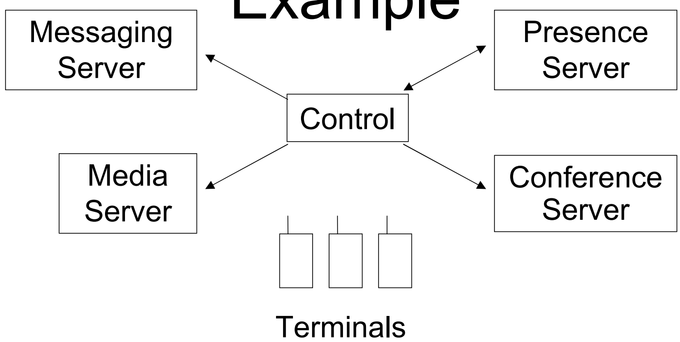
- Powerful tool e.g. for inviting users to centralized conferences or sessions with a Media Server
- In principle third party call control that has never been properly implemented in CSN, is as natural in SIP as first party call control because SIP is used also on the the interface to Application servers.

REFER and Call Transfer



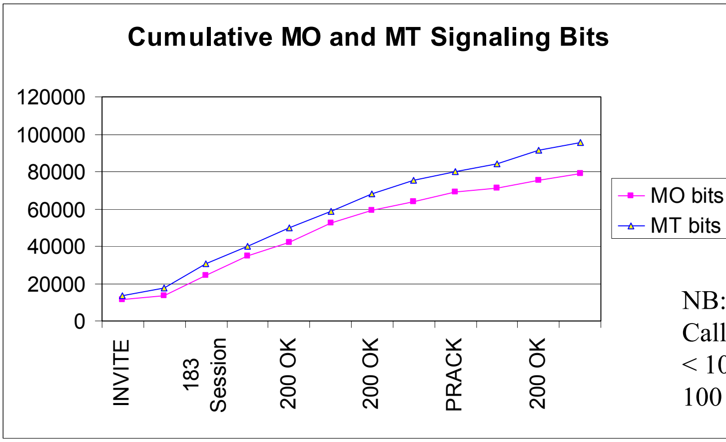
Media can always go directly from Transferee to Transfer target.

Auto-conferencing Service Example



1. One user orders the conference by filling a web form
2. Controller subscribes to each participants presence
3. When all available, send message or start IVR session to each participant to confirm willingness
4. Connect each participant to conference server. Play announcements to conference from media server when new parties join

Text based Signaling in IMS produces a lot of bits to the air interface in a cellular network



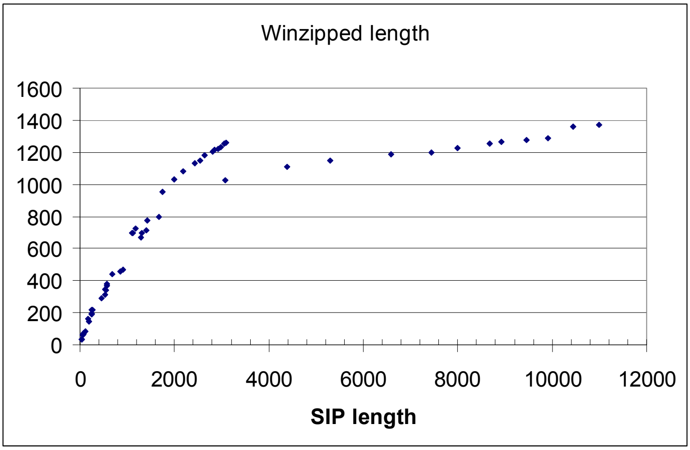
Cmp to ISDN

NB: length of DSS1(L2+L3) Call setup and release < 1000 bits→ SIP/SDP uses 100 times more bits

This is based on Release 5 MO session setup procedures from 24.228

+ The MO flow will create as many air interface bits as talking for 18 seconds using AMR 4.75 codec.
+ About 70% of the bits are created by redundant lines of text.
+ There is more redundancy on information element level

Zipping analysis of MO flow content shows that SIP/SDP carries a lot of context information in each transaction



Lengths are in bytes. The dots are created by applying WinZip to all incrementally longer subflows of the MO flow. The curve shows that by keeping state, we would save quite a lot in signaling bits.

Technical Problems

- How to make service components really independent?
- If there is dependency, how to move parameters between the components?
- How to secure call release?
- Signaling efficiency for narrow band services
  - Problem for narrow band networks and for shared capacity networks when SIP applied to narrow band services
  - SigComp gives some relief with the expense of cpu cycles and memory (most likely less than 1:10 compression)
- Emergency calling in VOIP and IMS.

Emergency calls in IMS

- Requirements
  - different countries have different requirements and different numbers for Emergency calls (Europe 112, USA 911, Japan 119 etc)
  - US: mobile terminal has to be geographically located
  - Europe: the network has to place the call even if there is no SIM card. Call has to be routed to the right Emergency Center.
- IMS issues:
  - GPRS always authenticates the user.
  - Different numbers in different countries → routing problem for roaming customers
- IMS solution in Release 5: The terminal has to place the emergency call using the CS domain in 3G → all voice terminals have to support CS services. P-CSCF has to detect an incoming emergency call by a roaming customer irrespective in which country the customer is roaming and even if the P-CSCF is located in the home network.

Emergency calls in VOIP

- Requirement: The Emergency Center has to see the address of the caller to the emergency number.
- In PSTN the telehone extension has a location number that identifies the copper wire to the residence. The directory number of the caller can always be mapped to the location number and the address of the caller retrieved from a subscriber database.
- IP networks do not support location numbers. IP addresses are allocated to users dynamically. If the user is calling from home, the home address can somehow be identified from a DB. If the user is connected while away from home, VOIP may give a wrong address to the Emergency Center.

Business problems

- Broadband + VOIP will kill PSTN, this is painful for Incumbent Operators. There is no incentive to deploy VOIP aggressively.
- At the same time voice is becoming mobile.
  - e.g with very conservative mobile policy, ca 90% of call costs are incurred by mobile services in Universities and Politechnics in Finland.
  - Many people have little faith in any wireline voice service.
- How to retain control over Subscribers that have BB connection. Any third party can provide VOIP (with QoS problems not solved).
- Why would Mobile Operators deploy IMS and SIP for voice services when the CS subsystems provides all the needed voice services?
  - it may be that IMS will first be used for services other than VOIP.

Voting for VOIP

- Vendors have stopped developing CS telephony.
- BB deployment is proceeding: Examples of South-Korea, US.
- With wide spread BB, if operators do not deploy VOIP, someone will (e.g. SKYPE).

OECD Broadband Statistics to June 2006

Over the past year, the number of broadband subscribers in the OECD increased 33% from 136 million in June 2005 to 181 million in June 2006.

Fibre to the home is becoming increasingly important for broadband access, particularly in countries with high broadband penetration. **Japan** leads the OECD in fibre-to-the-premises (FTTP) with **6.3 million fibre subscribers** in June 2006. **Fibre subscribers** alone in Japan **outnumber total broadband subscribers in 22 of the 30 OECD countries**.

The total number of ADSL subscriptions in **Korea and Japan** have continued to decline as more users **upgrade** to fibre-based connections. **DSL** continues to be the **leading platform** in 28 OECD countries. Cable modem subscribers outnumber DSL in Canada and the United States.

The breakdown of broadband technologies in June 2006 is as follows:

DSL: 63%	Cable modem	Other technologies (e.g. satellite, fibre and fixed wireless)
63%	29%	8%

Broadband subscribers per 100 inhabitants, by technology, June 2006

	DSL	**Cable**	**Other**	**Total**	**Rank**	**Total Subscribers**
Denmark	17.4	9.0	2.8	29.3	1	1 590 539
Netherlands	17.2	11.1	0.5	28.8	2	4 705 829
Iceland	26.5	0.0	0.7	27.3	3	80 672
Korea	13.2	8.8	4.5	26.4	4	12 770 911
Switzerland	16.9	9.0	0.4	26.2	5	1 945 358
Finland	21.7	3.1	0.2	25.0	6	1 309 800
Norway	20.4	3.8	0.4	24.6	7	1 137 697
Sweden*	14.4	4.3	4.0	22.7	8	2 046 222
Canada	10.8	11.5	0.1	22.4	9	7 161 872
United Kingdom	14.6	4.9	0.0	19.4	10	11 622 929
United States	8.0	9.8	1.4	19.2	12	56 502 351
Japan	11.3	2.7	4.9	19.0	13	24 217 012
OECD	**9.7**	**4.6**	**1.2**	**15.5**		**180 866 265**

China (ca 50M BB connections), other Asian tigers, etc.

BB penetration according to DSL Forum:
Telecommunications International, Jan 2007

Access Technology	Subscribers (M)	Percent	Sept-2006
DSL	173	66%	
CaTV	60	23%	
FTTx	27	10%	
Other	3,4	1%	
Total	264		

NB: This is for the whole world!

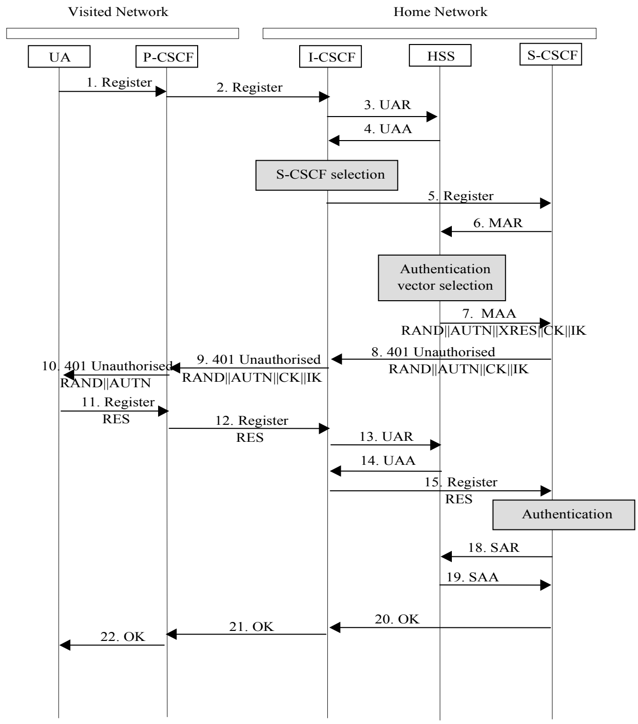
There is a very wide playing ground for session based services for BB users.

Conclusions on SIP

- SIP is a native IP-network signaling system suitable for Broadband networks
  - Needs compression when used e.g. in shared media cellular networks such as 3G WCDMA
  - Also, cellular networks of the future are going to be BB networks
- Most signaling and service architecture development in the world now is SIP oriented
  - Several IETF groups are producing a broad set of documents related to SIP
  - SIP architecture = base protocol + extensions
  - Newest developments include conferencing, Peer-to-Peer SIP etc.
- Deployment
  - BT NGN is based on SIP and IMS and will replace BT's PSTN in a few years
  - No attractive services in cellular networks so far based on IMS. Due to well working CS services, operators are not in a hurry to replace CS services with packet based IMS produced services in cellular networks. New attractive services are needed.

Appendix B – 3GPP IMS call flows

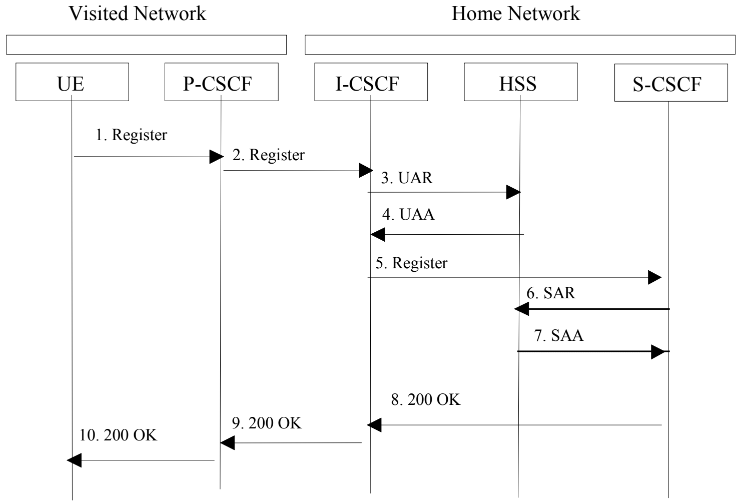
Registration – user not registered



Source: 29.228 v 7.0.0

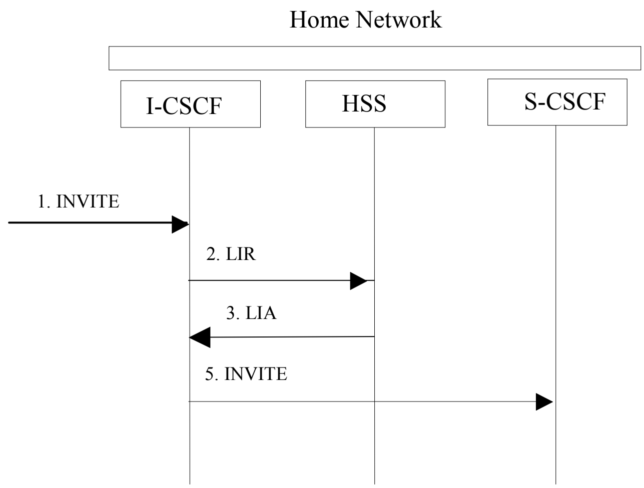
Registration – user currently registered

Source: 29.228 v 7.0.0

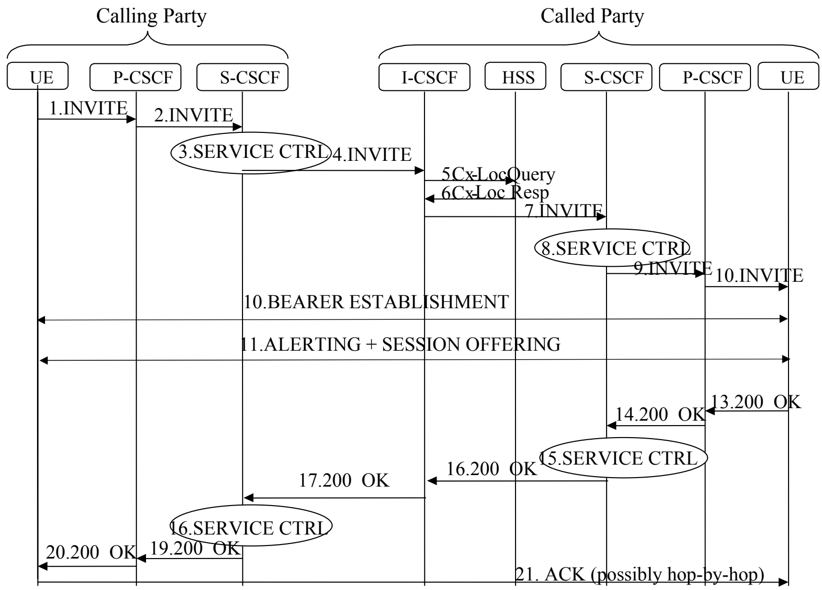


Mobile Terminated Session Setup

Source: 29.228 v 7.0.0



Mobile to Mobile Call



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The UE sends a INVITE (session destination) to the P-CSCF.

The P-CSCF forwards the INVITE (session destination) to the next hop name/address. In this case the next hop address is the S-CSCF.

The S-CSCF can read the information on who originated the INV, and forwarded this based on the session destination. The S-CSCF forwards the invite message to the I-CSCF.

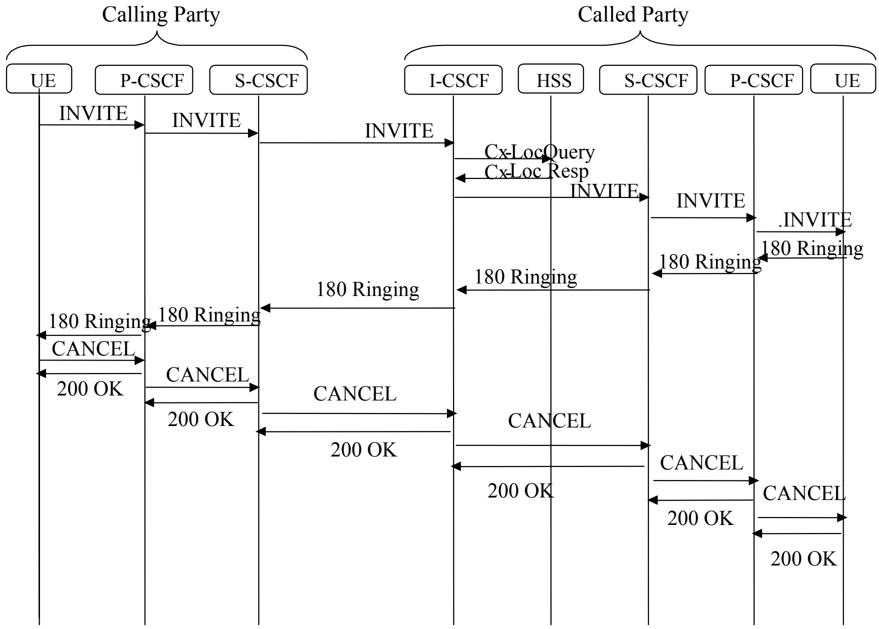
The I-CSCF sends ‘Cx-Location Query’ to the HSS to obtain the identity of the of the next hop which in this case is the S-CSCF.

The HSS sends the ‘Cx-Location Query Response’ to the I-CSCF.

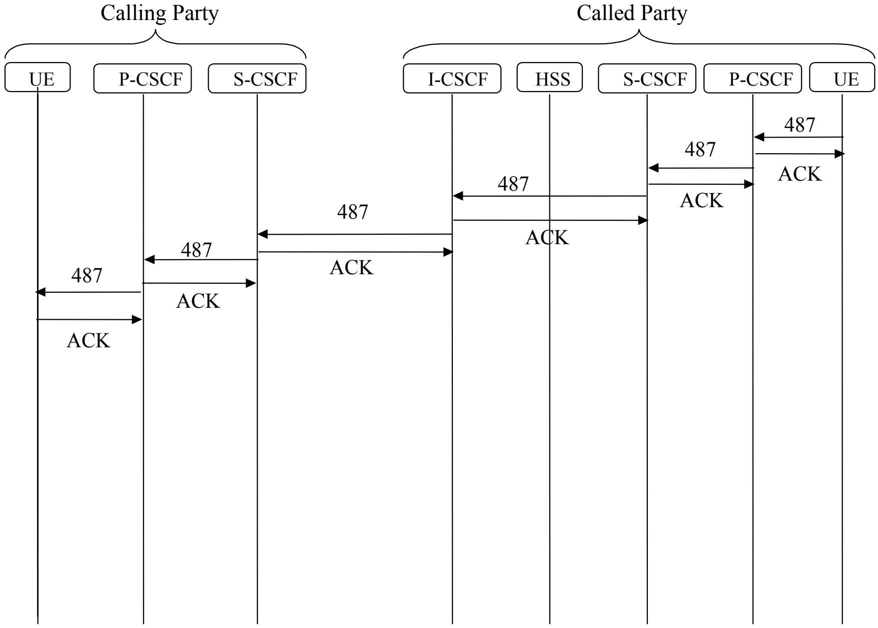
The I-CSCF forwards the INVITE message to the S-CSCF for the called party.

The S-CSCF carries out Service Control for Called Party. This includes applying the filter criteria to the incoming call and retargeting the the INVITE request. The latter means modifying the request-URI. The original URI is placed in the P-Called-Party-ID header field so the callee terminal will know to which of its public identities the call was addressed. Retargeting is necessary because the called party may be moving and thus keeping the original request-URI might create a loop.

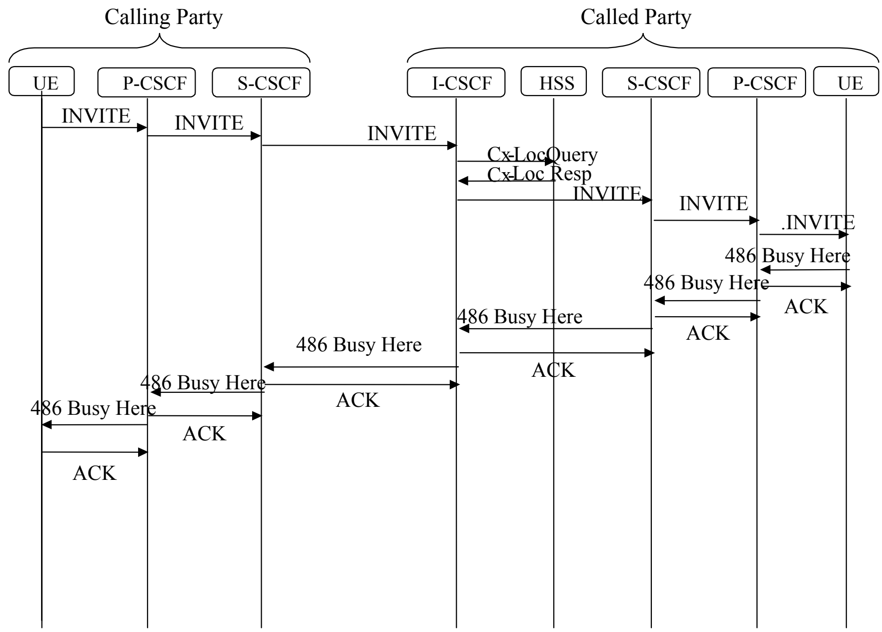
Call flow examples 1. - no answer



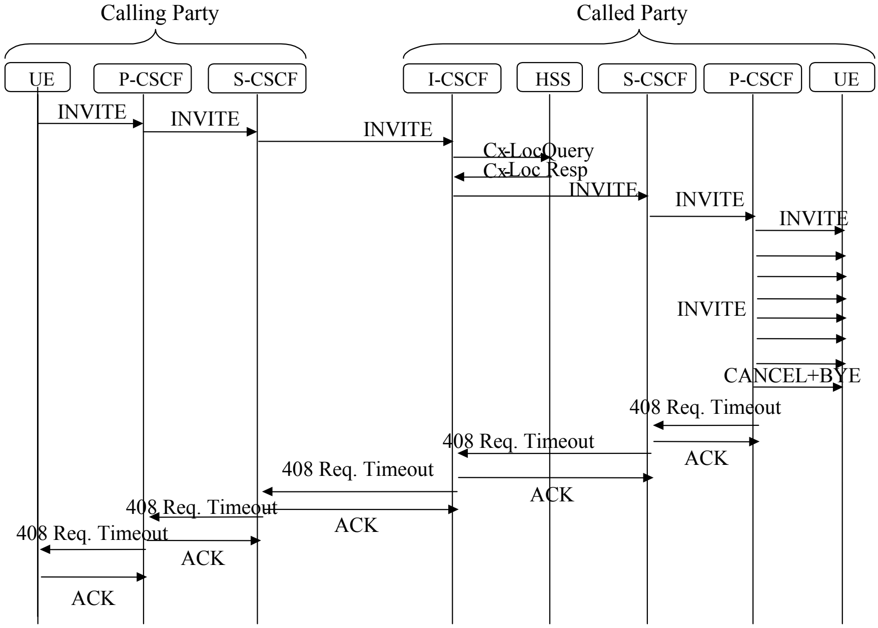
Call flow examples 1. - no answer 2.



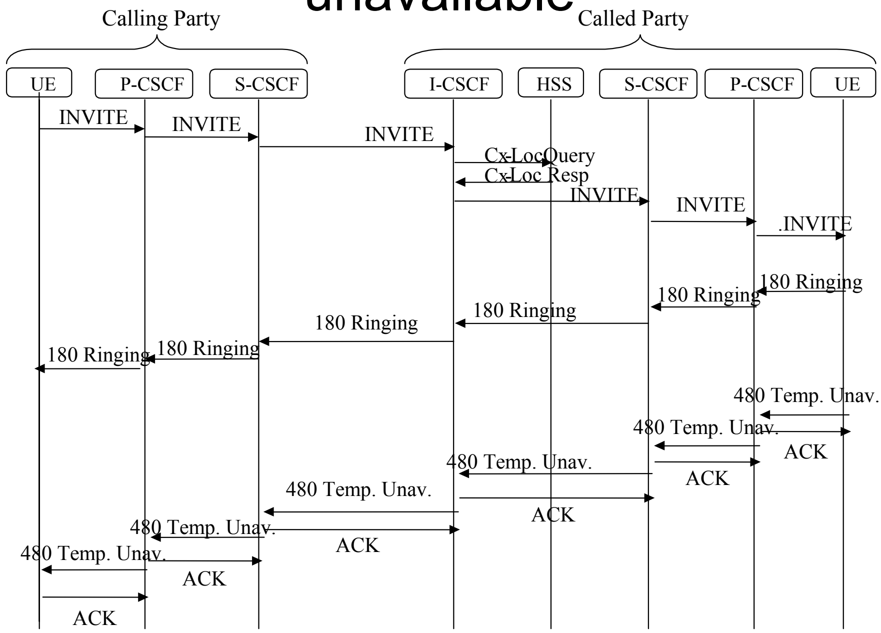
Call flow examples 2. - busy



Call flow examples 3. - no response

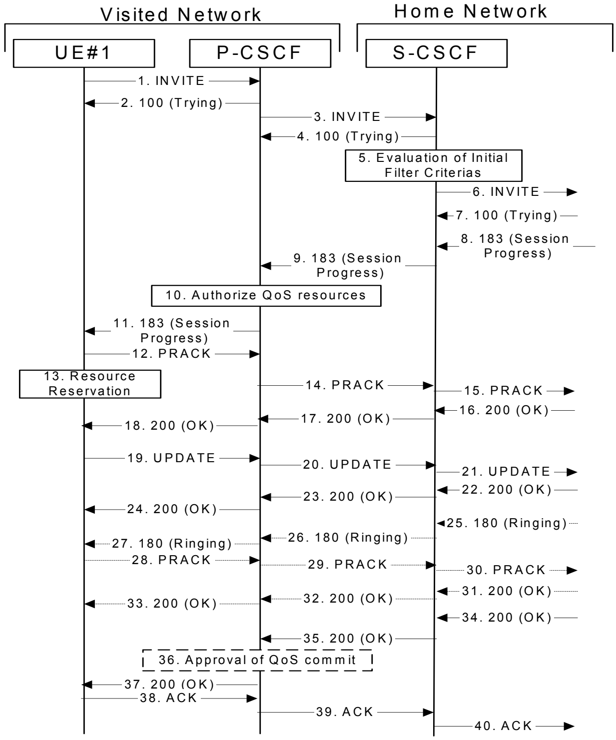


Call flow examples 4. - temporarily unavailable



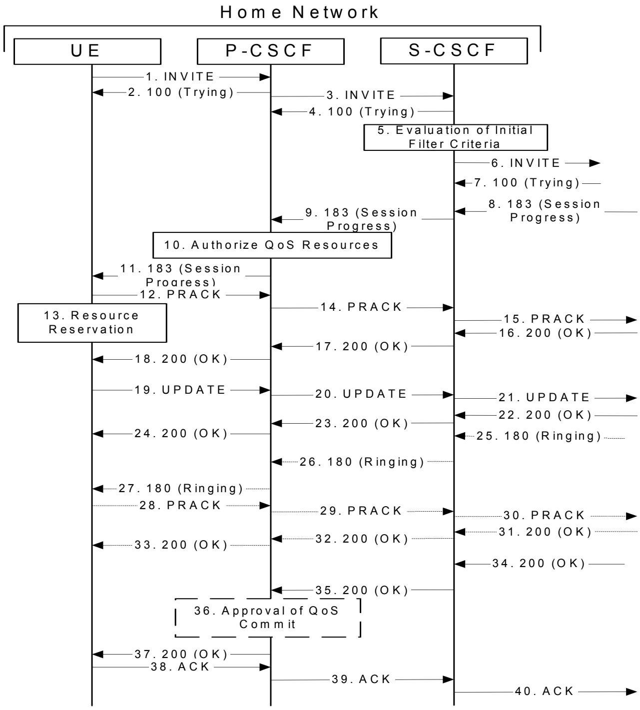
MO session setup - roaming

24.228 v 5.1.4

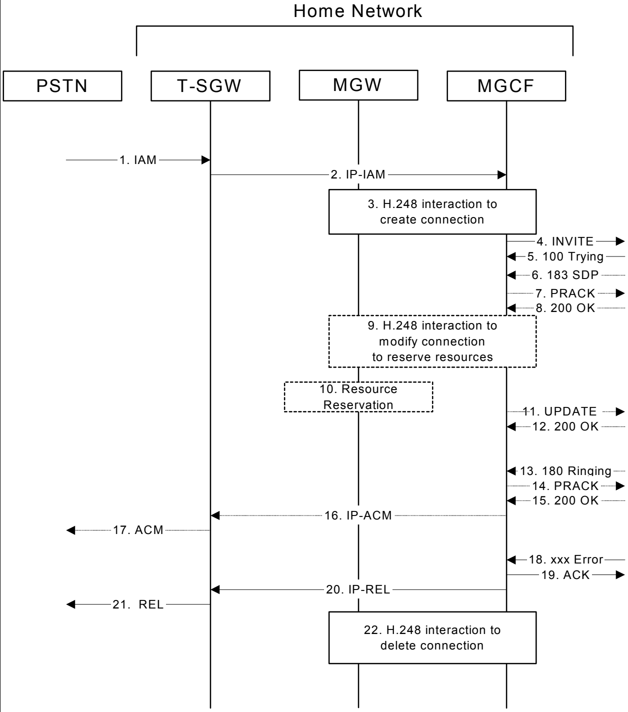


MO Session setup – user in home network

24.228 v 5.1.4



PSTN originated session



24.228 v 5.1.4